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their competitors; create new revenue sources; and/or expand existing revenue sources.

In order to provide enhanced telephone services, many telephone companies now implement a telephone communications network as an Advanced Intelligent Network (AIN) which has made it easier to provide a wide array of previously unavailable voice grade telephone service features. In an AIN system, telephone central offices, each of which serves as a signal switching point (SSP), detect one of a number of call processing events identified as AIN "triggers". An SSP which detects a trigger suspends processing of the call which activated the trigger, compiles a call data message and forwards that message via a common channel interoffice signaling (CCIS) link to a database system, such as a Service Control Point (SCP). The SCP may be implemented as part of an integrated service control point (ISCP). If needed, the SCP can instruct the central office (SSP) at which the AIN trigger was activated to obtain and forward additional information, e.g., information relating to the call. Once sufficient information about the call has reached the ISCP, the ISCP accesses stored call processing information or records (CPRs) to generate from the received message data, a call control message. The call control message is then used to instruct the central office on how to process the call which activated the AIN trigger. As part of the call control message, an ISCP can instruct the central office

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incorporated by reference, describes in detail a Wide Area Centrex system implemented using AIN techniques.

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Voice dialing is a useful service which has  
5 been implemented in some known systems by having control  
logic in a telephone switch connect a caller to a voice  
dialing IP which provides the voice dialing service.  
The telephone switch may couple the subscriber to the  
voice dialing IP as a result of the subscriber calling a  
10 telephone number corresponding to the IP or entering a  
code which is detected by the switch. Such known voice  
dialing systems are not AIN based and therefore are  
somewhat limited in terms of the logic and information  
available for controlling connections to voice dialing  
15 apparatus, e.g., voice dialing IPs. Thus, the known  
techniques of using logic embedded in a switch to  
determine when and to which voice dialing IP a caller  
should be connected can lead to inefficient using of  
voice dialing IP resources and limit the ability of known  
20 voice dialing services to be implemented as an integral  
part of other services, e.g., Centrex Services.

Voice dialing is a particularly desirable  
service since it eliminates the requirement that a user  
25 of the voice dialing service remember the telephone  
number of the party being called. In various known voice  
dialing systems, speaker dependent speech recognition is  
used to identify spoken names. In such systems, a  
personal dialing directory is maintained for each

subscriber of the voice dialing system. The personal dialing directory is a database which includes a speaker dependent speech recognition template for each of a plurality of names which may be spoken and a telephone number for each name. When used, the data, e.g., templates in a subscriber's directory, are retrieved; a speaker dependent speech recognition operation is performed on a spoken name provided by the subscriber; and then, assuming a name is recognized, the call is completed to the telephone number in the subscriber's personal dialing directory corresponding to the name. An IP may be used for performing the speech recognition and various other tasks associated with the known voice dialing operation. One particular system for implementing voice dialing is described in detail in U.S. Patent No. 5,832,063.

While voice dialing systems which use speaker dependent speech recognition to identify spoken names enjoy a high degree of recognition accuracy, they have the disadvantage of requiring that a user of the system provide one or more utterances of each name for which speaker dependent speech recognition templates are to be generated. Thus, a voice signal, e.g., voice telephone connection, is normally required when adding or updating names in a personal dialing directory. In addition, the need to provide multiple utterances of each name in the personal dialing directory can prove irritating to some customers.

Unfortunately, the use of speaker dependent speech recognition templates, with the corresponding need for multiple speech samples to train each name in a personal dialing directory, has made Internet based management of existing voice dialing systems of the type described above difficult to implement.

While existing Centrex and voice dialing services are useful, it is desirable that such services continue to be improved and enhanced. With regard to voice dialing, it is desirable that new methods and apparatus be devised which would allow for Internet based management of voice dialing services. It is also desirable that new methods of providing voice dialing services be devised which will allow voice dialing services to be implemented as part of AIN based service packages such a Centrex. With regard to Centrex, it is desirable that Centrex service be enhanced to support

voice dialing functionality as well as Internet based management of said functionality.

5 SUMMARY OF THE INVENTION

The present invention is directed to methods and apparatus which can be used to provide voice dialing, enhanced Centrex, and other communications services. In various embodiments, the communications services of the present invention are implemented to facilitate easy management of the services by end users via the Internet. In accordance with one feature of the present invention, voice dialing is implemented as an AIN based service. By implementing voice dialing as an AIN based service, the service can be easily integrated with Centrex and/or other AIN based services. In addition, since control logic in an ISCP is used to control access to voice dialing hardware, in response to activation of one or more AIN triggers set telephone switches, access to voice dialing hardware can be controlled to provide efficient use of the voice dialing hardware regardless of the location from which a voice dialing service subscriber calls. In addition, ISCP logic can translate multiple identifiers, e.g., various telephone numbers, associated with a subscriber, into a single user identifier which can be used by voice dialing hardware for subscriber information retrieval purposes.

In accordance with the present invention, a voice dialing call may be placed to an individual or party by speaking either the name or a nickname of the individual or party being called. In addition, if desired, a location associated with the party or individual may also be specified in addition to the party or individuals name. When a location is stated in conjunction with a name, the call will be placed to the party or individual at the particular specified location.

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A subscriber's voice dialing record is stored  
25 in part of the public telephone system in accordance with  
the present invention, e.g., in an intelligent peripheral  
device, e.g., voice dialing (VD) IP, coupled to a central  
office telephone switch. In addition to being coupled to  
the telephone network via the central office switch, the

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Speaker independent speech recognition models produced from text work well for most names. However, in some cases where the pronunciation of a name is difficult to predict from its spelling, e.g., due to a variety of possible pronunciations or because it originates from a foreign language name, a speaker independent speech recognition model produced from text may provide recognition results which are less than satisfactory.

In order to address this potential problem, the methods and apparatus of the present invention allow for

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a subscriber to provide one or more speech samples of a name to be used for generating a speech recognition model for voice dialing purposes. The speech corresponding to a name may be provided as part of a telephone based voice dialing entry creation or updating process. In the case where an existing entry is being updated using a spoken version of a name, a speaker independent (or, optionally speaker dependent) speech recognition model is generated from the speech corresponding to the name. The generated speech recognition model is stored in place of a speech recognition model previously generated from text when such a text based model exists.

In the case where speech corresponding to a name is being used to create a voice dialing entry, as opposed to update an existing entry, the speech corresponding to the name is used to generate a speech recognition model. A speech recognition operation is also performed on the speech and a text version of the spoken name is generated and used to populate the (text) name portion of the calling entry corresponding to the spoken name. The text name entry is displayed to a user accessing the calling entry information via the Internet. A user can edit, via the Internet, the spelling of the name in the entry if desired without affecting the speech recognition model generated from the supplied speech.

To distinguish between speech recognition models generated from speech and those generated from

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Upon recognizing a name in received speech, the voice dialing IP plays the voice dialing subscriber a confirmation message such as, e.g., "Dialing John Smith" where "John Smith" is the recognized name. This provides the subscriber the opportunity to stop the call in the event a recognition error occurred, e.g., by hanging up or otherwise signaling the voice dialing IP. A voice recording of the recognized name may be played to the subscriber as part of the confirmation message, e.g., recording of the subscriber's voice obtained at the time the subscriber trained a speech recognition model in his/her voice dialing directory using speech. However, in accordance with one feature of the present invention, a text representation audio corresponding to a recognized name is generated for message confirmation purposes from a stored text version of the name. In such an embodiment, a text to speech circuit is used to generate an audio version of a name from a stored text version when an audio version of the name is needed for a confirmation message. Since storage of text is much more efficient than the storage of voice recordings, the use of text versions of names for confirmation message purposes can be considerably more efficient from a memory perspective than the use of audio recordings.





Thus, the present invention provides a voice dialing subscriber the opportunity to place multiple sequential voice dialing calls without having to hang-up when a busy signal or no answer condition is encountered. Because the voice dialing IP is disconnected from the caller between voice dialing call attempts, voice dialing and speech recognition resources are used in an efficient manner and are not tied up during the time required to detect the outcome of a call.

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BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 illustrates a communication system implemented in accordance with an exemplary embodiment of the present invention.

Fig. 2 illustrates an exemplary Centrex internet connection access server which may be used in the system illustrated in Fig. 1

Fig. 3 illustrates an exemplary voice dialing intelligent peripheral device suitable for use in the system of Fig. 1.

Fig. 4 is a detailed illustration of an exemplary speech recognizer array which can be used in the voice dialing IP of Fig. 3.

Fig. 5-9 are flow diagrams illustrating steps, messages, and data associated with setting up, configuring, and using Centrex account information in accordance with the present invention.

Figs. 10-14 illustrate various pages which may be displayed to a Centrex voice dialing service subscriber when the subscriber accesses and/or updates voice dialing service information via the Internet.

5                    Fig. 16 illustrates a sub-routine used for  
updating voice dialing entries based on information  
supplied over the Internet to the voice dialing IP of  
Fig. 1.

15                Figs. 18-20 are call flow diagrams which illustrate the steps, in addition to the data, instruction and message flow, associated with making one or more voice dialing calls in accordance with the present invention.

As discussed above, the present invention is directed to methods and apparatus for providing voice dialing services. The voice dialing services may be provided as a stand-alone service, as part of a Centrex service, or as part of another telephone service package. As will be discussed in detail below, in accordance with

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Each SSP 2, 4, 6 is normally connected to one or more customer premises (CP) which may include, e.g., Centrex subscriber residences and/or offices as well as

In the Fig. 1 example, first and second customer premises 22 and 24 are coupled to the second SSP 4, while third and fourth customer premises 26, 28 are coupled to the third SSP 6. Connections between the SSPs and CPs may be by POTS lines, ISDN lines, DSL, or other known communications lines.

Communications equipment, referred to as customer premise equipment (CPE) is located at each customer premises 22, 24, 26, 28. Customer premise equipment may include, e.g., telephones, faxes, computers, etc. In Fig. 1, a computer 36, land-line telephone 38, and mobile telephone 37 are shown as being located at the first customer premises 22. As will be discussed below, each of these devices corresponds in the exemplary embodiment shown in Fig. 1 to a first Centrex service subscriber. Since cell phone 37 is a mobile communications device it need not be physically located at the first customer premises to operate. The computer 36, located at the first customer premises 22 is coupled by any one of a plurality of known connection techniques, e.g., telephone dial-up, ISDN, DSL, etc., to the Internet 30, also known as the World Wide Web.

While the second, third and fourth customer premises 26, 28 are illustrated as including only landline phones, it is to be understood that they may

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The ISCP 16 includes an SCP 64, a service management system (SMS) 62, data and reporting system (DRS) 63, service creation environment (SCE) 60, and a network interface (NI) 45. A local network 67 couples the various components of the ISCP 16 together.

The SCP 64 includes a multi-service application platform (MSAP) database 69 which includes customer data (CD) 71 for each of a plurality of Centrex and/or other service subscribers. The customer data 71 includes, for each customer: 1) a list of the services to which the customer subscribes; 2) a password which may be input via DTMF signals; and 3) a call processing record (CPR) which is used to instruct an SSP how to process a call in response to an AIN trigger to thereby implement the services to which the customer subscribes. Exemplary services which may be supported by the ISCP 16 include, e.g., voice dialing, call forwarding, call screening,

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voice mail and a host of other services which may be provided to Centrex subscribers as well as non-Centrex telephone customers.

5           For purposes of explaining the voice dialing features of the present invention, voice dialing services will be described in the context of a Centrex environment. However, it is to be understood that the voice dialing service of the present invention can be  
10 provided outside the Centrex environment, i.e., voice dialing services can be provided to non-Centrex telephone customers as well as Centrex service subscribers.

The customer data 71 which includes call processing records 73 is generated, at least initially, by the SCE 60 in response to input received from a service representative or operator 44. Customer data in the database 71 may, after initial provisioning of a service for a customer, the CPR may be updated by the customer via the Internet and the use of a Web browser.

The SCE 60 includes an operator terminal (OT) 49, service order processing circuitry 48 and AIN provisioning system circuitry 46. The operator terminal 49 is used by the service representative 44 to enter service information, e.g., to create a Centrex account for a new subscriber. The entered data may be information, e.g., relating to the addition of a new Centrex customer, the adding of a service for an existing



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detection operations as well as playing voice prompts and other messages to a Centrex customer.

DTMF IP 10 is coupled to the first SSP 2 via  
5 audio (voice) and signaling lines. It is also coupled to  
the OSN through a network interface (NI) 21. In this  
manner, the DTMF IP 10 can interact with other components  
of the system 100, e.g., ISCP 16, via communications  
transmitted over OSN 34 or through the SSP 2. The DTMF  
10 IP 10 may be implemented using known hardware.  
Accordingly, the hardware used to implement DTMF IP 10  
will not be described in detail.

The DTMF IP 10 serves as a platform by which a  
15 Centrex service subscriber can update his/her service  
information, e.g., voice dialing directory information,  
through a telephone as opposed to an Internet connection.  
A Centrex service subscriber can establish a service  
updating or management session with the DTMF IP 10, by  
20 dialing a telephone number associated with the DTMF IP  
10. Dialing of the DTMF's telephone number results in  
the subscriber's call being routed to SSP 2 and a  
voice/DTMF connection to the DTMF IP being established.

25                   DTMF IP 10 includes various security features,  
e.g., customer identification and password entry  
requirements, as does the CICAS 32, to insure that  
Centrex customers are limited to accessing and updating

their own service records and not those of other Centrex service subscribers.

The second and third IPs 18 and 20 are voice dialing (VD) IPs which are dedicated to supporting voice dialing services in accordance with the present invention. Each of the voice dialing IP's includes a network interface 19, 21 which couples the VD IPs to the OSN 34. The VD IP's are also coupled to the second and third SSPs 4, 6, respectively, via voice and signaling links. An exemplary VD IP will be discussed in greater detail below with regard to Figs. 3 and 4.

The CICAS 32 will now be discussed briefly with regard to Fig. 2. As illustrated, the CICAS 32 comprises first and second network interfaces 150, 152, a processor 154 and memory 156 which are coupled together as shown in Fig. 2. The first network interface 150 links the CICAS 32 to the Internet 30 while the second network interface 152 links the CICAS 32 to the OSN 34. Thus, the CICAS 32 serves as a gateway by which a service subscriber can gain access from the Internet, after being authenticated, to the various telephone system network components, e.g., the ISCP 16, and VD IPs 18, 20. Through use of the CICAS 32 Centrex service subscribers can manage the Centrex services to which they subscribe via their personal computers and a Web Browser application.

20           In addition to the Centrex subscriber  
information 159, memory 156 includes a set of Centrex  
management server routines.

routine 168. The routines which form the Centrex

providing a subscriber access to Centrex subscriber information and hardware required to manage the subscriber's Centrex service.

5           The Web access server routine 161 is responsible for initially interfacing with the Centrex service subscriber and for providing web pages to be displayed. The server routine 161 interfaces with the other routines, which are responsible for performing security checks, retrieving subscriber data, and performing other functions.

The transaction server routine 162 is responsible for determining what other routines need to be accessed to provide the user with a requested service or transaction.

The database server routine 166 is responsible for controlling access to and retrieval of locally stored information, e.g., Centrex information included in the subscriber information database 159. The database server routine 166 interacts with the other routines 162, 164, 168 to provide Centrex subscriber information as required.

The ICAS broker routine 164 is responsible for determining the availability of system components, e.g., system resources, to meet the needs of the various other routines including the transaction server routine 162 and

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In order to perform various operations, the IP 18 includes an application processor (CPU) 75 and memory 72. The memory includes a set of routines 73 and a database 85. The application processor 75 is coupled to the switching interface 74, network interface 19, memory 72, and speech recognizer arrays 70, 71 via a local bus 79 which couples the components of the voice dialing IP together.

The database 85 includes information used by the IP 18 to perform voice dialing operations as well as



The application processor 75 executes the various routines 73 when called upon to perform various functions supported by the voice dialing IP 18. Voice dialing Web server routine 76 is responsible for presenting a Centrex voice dialing service subscriber the voice dialing information included in the subscriber personal dialer record via OSN 34 and the CICAS 32, e.g., in an HTML format that can be displayed using a web browser. The Web server routine 76 is also responsible for receiving from the subscriber data representing modifications to the subscriber's voice dialing record and for calling the Internet based personal dialer updating routine 82 to control the updating of the subscriber's personal dialer record based on the information received from the customer via the Internet.

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During voice dialing operations, speech recognition models, e.g., speaker independent speech recognition models, stored in a subscriber's voice dialing customer record 88, 90 are retrieved from the database 85 and supplied to speech recognition circuit 104, 106 or 108 assigned to service the call. The retrieved models may be used in conjunction with speaker independent models of commands, names of locations, and/or numbers. The speaker independent speech recognition models obtained from the customer record 88, 90, which may be in the form of phonetic representations of the modeled names, are used to determine which, if any, name in a subscriber's personal voice dialing

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directory was spoken. The other speaker independent models are used during a voice dialing operation to detect spoken commands, names of locations and/or numbers, e.g., digits to dial or numbers spoken in response to a prompt for input.

In one embodiment, during a voice dialing operation, recognition of phones, or phonemes which maybe used synonymously, is performed in an integrated way as part of the overall speech recognition operation. As part of the speech recognition operation, the incoming speech, after appropriate processing, is compared against a network of possible phone/phoneme sequences dictated by the speech recognition models used and associated grammar. The best path in the network, corresponding to a particular word, phrase or result, e.g., no valid word recognized, is selected during the speech recognition operation and corresponding word, phrase or result is declared as the outcome of the speech recognition operation. Accordingly, in at least one exemplary voice dialing implementation, phones are recognized in the context of the expected phone-strings, dictated by a grammar that defines the space of possible text strings, to be spoken by the caller, for that application, e.g., voice dialing.

When performing a voice dialing operation, in the event that 1) a name from the subscriber's personal dialing directory is not recognized, and 2) a telephone

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The components of the speech recognizer arrays 70, 71 operate in conjunction with the application

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In step A1 the person wishing to subscribe to Centrex service places a call to a Centrex service subscription number. The call is received by the SSP 4 to which the caller is coupled. Thus, in the example the second SSP 4 becomes the originating SSP. In step A2, the originating SSP 4 sends an initial address message (IAM) to the SSP 2 corresponding to the dialed Centrex service subscription number. Thus, the SSP to which the Centrex customer service representative (CCSR) 44 is connected becomes the terminating SSP for this portion of the processing.

In step A3, the service representative responds to the caller and prompts the caller for subscription information including the landline and mobile telephone numbers for which Centrex service is to be provided. Then in step A4, the caller provides subscription information, including the requested telephone numbers,



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At the end of the process illustrated in Fig. 6, in exit state <B\_00e>, it can be seen that one or more Centrex services are associated with the new subscriber's lines and that the CPR for the new subscriber includes  
5 the subscriber's temporary PIN value.

Fig. 7 illustrates the process by which a new Centrex subscriber activates his/her Centrex service by dialing a number corresponding to the Centrex DTMF IP 10,  
10 for the first time. In Fig. 7, the initial state <C-00b> is the same as the Fig. 6 exit state <B\_00e>. Accordingly, in state <C-00b> one or more Centrex services have already been associated with the new subscriber's lines and the new subscriber's CPR includes  
15 a temporary PIN value.

To activate the Centrex service, in step C1, the user places a call from the subscribing line, e.g., phone 38, to the Centrex service activation number  
20 previously provided. Since the dialed Centrex activation telephone number corresponds to Centrex DTMF IP 10, the SSP 4 which, in this example is the originating SSP, sends, in step C2, an initial address message (IAM) to SSP 2 to which DTMF IP 10 is connected. Thus, in this  
25 example, SSP 2 is the terminating SSP. In step C3, termination SSP 2 then forwards the new subscriber's call to the Centrex DTMF IP 10 to which the call was directed.

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10           The process illustrated in Fig. 8 begins in  
step D1 wherein the subscriber submits, using a Web  
Browser application, a request to the CICAS 32 for access  
to a Centrex Web access default web page. The request is  
forwarded through the network interface 150 to the Web  
5   access server routine 161. Next, in step D2, the Web  
access server routine returns to the subscriber a Web  
Access default page 100, e.g., a page of the type  
illustrated in Fig. 10. The subscriber's Web Browser  
displays the default page on the screen of computer 36.

Next, the subscriber requests to manage his/her existing Centrex account by, e.g., clicking on the displayed "MANAGE YOUR EXISTING CENTREX ACCOUNT" option. This causes, in step D3, the subscriber's Web browser application to send a Centrex service access request to the Web access server routine 161. In response to the service access request, in step D4, the Web access server routine 161 initiates a secure communications session with the subscriber's Web browser and then sends the

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In step D17, the ICAS security server routine 168 saves the submitted long password and other service information, e.g. in the local set of Centrex subscriber information 159, and then signals the subscriber that the set-up operation has been successfully completed.

Accordingly, at the end of the process illustrated in Fig. 8, the new subscriber is

authenticated, a long password to be used for Internet access to Centrex services exists for the subscriber and one or more Centrex services are authorized for the subscriber.

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The process of accessing and updating Centrex subscriber account information via the Internet 30, after an initial Internet access operation, is illustrated in Fig. 9. In Fig. 9, the initial state <E-00b> is the same as the Fig. 8 exit state <D\_00e>. Accordingly, in state <E-00b> the subscriber's long PIN used for Internet access and other account information has already been stored in the set of Centrex subscriber information 159 maintained by the CICAS 32.

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In step E1, a customer request to access his/her Centrex account is received by the Web access server routine via the Internet 30 and the network interface 150. In step E2, the Web Access Server Routine returns the default web page 1000 to the subscriber. In step E3, the subscriber requests to manage his/her existing Centrex account. Then, in step E4, the Web access server routine 161 initiates a secure session with the subscriber's Web browser application and requests Calling Party Id and PIN (password) information, e.g., by sending the page 1100 to the subscriber's browser.

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In step E5, the user provides the requested ID and password information. Then in step E6, Web access

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In step E8, assuming the subscriber was authenticated, the Web access server routine 161 presents the subscriber with a Centrex management service selection page which lists options for managing the various services to which the user subscribes. Fig. 12 illustrates an exemplary Centrex management service selection page 1200 wherein the subscriber is presented with the option of managing his/her voice dialing service, call forwarding service, call screening service, account/password information, or voice mail.

In step E9, the user communicates the selected management operation to the Web access server routine 161. Then in step E9, the Web access server retrieves the Centrex subscriber information it has available to it from its own database relating to the selected service and, if another device e.g., IP is responsible for maintaining customer records relating to the selected

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The Internet Based personal dialer updating routine 82 is activated by the voice dialing web server routine in response to the subscriber's request to manage/update his or her personal voice dialing service.

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The routine 82 begins by displaying via the user's Web Browser, Voice Dialing window 1300 illustrated in Fig. 13. The window displays voice dialing information included in the subscriber's voice dialing record which the user may wish to modify, add to, or delete. It also displays control options, e.g., options such as the option of adding a new entry 1308, and modifying the subscriber's password, which the user can select, e.g. by double clicking on the displayed option.

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The voice dialer information displayed to the subscriber includes, e.g., the subscriber's Centrex phone number 1302, the subscriber's mobile phone number 1304, the subscriber's easy access number 1306, the name of a corporate dialer 1320 if the subscriber has designated one to be used in conjunction with the subscriber's personal dialer, and voice dialing entries, e.g., calling records 1310, 1312, 1314, 1316. Each calling record corresponds to one individual or party who the voice dialing subscriber can call through the use of voice dialing, e.g., by speaking the person's name or nickname.

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Unlike some known voice dialing systems, the voice dialing system of the present invention allows

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used. If only a name is detected, it is assumed that the first telephone number corresponding to the name is to be used for dialing purposes.

5           Locations which may be specified can be  
selected from the following list of identifiers:

	OFFICE	OFFICE 1	Office 2	Office 3
	OFFICE 4	Second Office	Branch Office	Headquarters
10	Secretary	Home	Home 1	Residence
	Residence 1	Residence 2	Residence 3	Mobile
	Mobile 1	Mobile 2	Mobile 3	Temporary Location

15           By limiting the list of locations which can be  
designated, the need for the VD IP to store speech  
recognition models for custom locations is avoided and a  
single set of speaker independent speech recognition  
models can be used for detecting the spoken name of a  
20 location for multiple subscribers. For example, to  
support the above list of 20 possible locations, 20  
speaker independent speech recognition models are stored  
by the VD IP 18 and multiple subscribers.

25           When presented with the voice dialer information 1300, the subscriber can choose to modify the displayed information, add a new entry or to change his/her password, e.g. by double clicking on an existing entry, icon 1308, or icon 1309, respectively. A

subscriber's selections are communicated by the subscriber's computer to the VD IP 18 via the Internet.

If the subscriber selects to add a new entry or  
5 to edit an existing entry, the subscriber is presented  
with a voice dialing entry modification page 1400 which  
is displayed by his/her browser. The page 1400 includes  
information relating to the entry to be added or  
modified.

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While having much of the same entry information as the voice dialer page 1300, the voice dialing entry modification page includes additional information relating to an entry as well as entry modification information, e.g., an icon 1407 which may be selected for display a listed location names from which the subscriber can choose when editing the contents of an entry. In addition, the page 1400 includes a submit updated entry icon 1410. The icon 1410 may be selected by the user, e.g., by double clicking, to submit the entry information to the VD IP 18 after the user makes the desired changes to the entry. In response to selection of the submit updated entry icon 1410, the user's web browser transmits the entry information, as modified by the user, to the VD IP 18.

In Fig. 14, the first column 1401, of entry 1312, is a name entry. In the second column 1402, model type information is displayed. As discussed above,

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spoken location is not detected, the default telephone number associated with the detected name or nick-name is used to route the call.

5           The subscriber can edit the information shown  
in the columns 1401, 1402, 1403, 1404, 1405, 1406 as  
desired. Upon completing the modifications to the entry  
1312, the user selects the submit updated entry icon 1410  
and the revised entry information is transmitted via the  
10 Internet to the VD IP 18. As will be discussed in detail  
below, the VD IP 18 updated the calling record  
corresponding to the entry as may be required by the  
user's changes.

15           While an existing entry 1312 is shown in Fig.  
14, a new entry would be displayed in a similar fashion  
with the information in each of the six columns being  
blank. The user need not fill in columns 1402, 1404  
since speech recognition models for new entries created  
20 via the Internet updating process are generated from the  
text name with columns 1402 and 1404 then being updated  
VD IP 18 accordingly. The speech recognition model  
generated from the supplied text name will later be  
replaced with a voice trained model should recognition  
25 accuracy achieved with the speech recognition model  
generated from text prove unsatisfactory to the  
subscriber.

Fig. 15 illustrates an exemplary voice dialing customer record 1500 which is stored by the VD IP 18 for a voice dialing subscriber. The record 1500 may be used as any one of the customer records 88, 90 illustrated in Fig. 3. One voice dialing record 1500 is stored for each subscriber. As illustrated, the record comprises much of the same information that was shown in Figs. 13 and 14. A ' mark is added to each of the numbers in Fig. 15, which corresponds to previously discussed data shown in figs. 13 and 14 to distinguish between the stored data in the record 1500 and the displayed data of Figs. 13 and 14.

Note that in addition to the data previously discussed with regard to Figs. 13 and 14, the customer record 1500 includes the subscriber's Internet PIN 1501 and the speech recognition models, SI MODELS 1-7, shown in columns 1502 and 1504. In the illustrated embodiment, one model speaker independent speech recognition model is stored for each name and nickname included in the customer's record.

As discussed above, speech recognition models may be generated from a speech sample provided by the subscriber, e.g., over a voice telephone connection, or from text provided via the Internet. When dialing entries are initially generated from speech, phoneme to text conversion is used to generate a text name corresponding to a generated speech recognition model. In addition, the model type identifier, MT, for the

generated model, is set to S indicating the model was generated from speech. Often the text spelling of the name may include errors resulting from the phoneme to text conversion process. In accordance with the present invention, a subscriber can update the text name, via the Internet, without affecting the corresponding speech model. This can be done by modifying the text of the name but leaving the model type identifier set to S. In accordance with the present invention, this results in the text name information being updated, e.g., corrected, with the speech recognition model being left unaltered. Should a subscriber wish to replace a speech recognition model generated from speech with one from text, the subscriber can change the "S" in the model type field to a "T". This will result in a speech recognition model being generated from the text of the name. The generated model will be stored in place of the previously generated speech recognition model. Thus, when the text of a name is altered, the text entry is updated. However, when there is an exiting speech recognition model which was generated from speech (MT=S) it is replaced only if the user modifies the model type identifier to indicate the model is to be generated from text.

Fig. 16 illustrates voice dialing entry updating sub-routine 1600 which is part of the Internet based personal dialer updating routine 82. The entry updating sub-routine 1600 begins in step 1602 when it is executed by the VD IP 18 in response to receiving updated

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If an S exists in the MT field it indicates that the existing speech recognition model is not to be altered and operation proceeds to step 1616.

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field value was changed from S to T operation proceeds to step 1612. Otherwise operation proceeds to step 1610.

In step 1610, a determination is made as to whether the name was modified by the subscriber. This check can be made by comparing the previous value in the name field, i.e., the value supplied to the subscriber's web browser for display, to the value supplied by the web browser to the VD IP 18 as part of the modified entry information. If the name was not modified, updating to the speech recognition model is not required and operation proceeds directly to step 1616. If, however, in step 1610 it is determined that the name was modified, operation proceeds to step 1612.

In step 1612, a speech recognition model is generated from the text of the name, obtained from the subscriber via the Internet. A text to phoneme operation is normally performed as part of the speaker independent model generation process. Text to phoneme information 98 may be used when converting the text information to phoneme information.

After the speaker independent speech  
25 recognition model is generated in step 1612, operation  
proceeds to step 1614 wherein the generated SI speech  
recognition model is stored in the customer record with  
the model being associated with the name from which it  
was generated. As part of the storing operation, a model

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In step 1626, the entry updating sub-routine is stopped pending its re-execution to processes another set of modified voice dialing entry information.

5           As discussed above, a voice dialing service subscriber can also update his/her voice dialing record through the use of an ordinary telephone. In such a case, the subscriber calls a telephone number assigned to servicing voice dialing customers. The subscriber is first connected to the Centrex DTMF IP 10 which collects the user's ID, e.g., Centrex telephone number and PIN. After the ISCP 16 validates the subscriber, assuming the subscriber's PIN is correct, the subscriber is connected to the VD IP 18. The phoneme based personal dialer updating routine 84 is invoked, and the subscriber's voice dialing record is retrieved based on the User ID (Centrex telephone number) obtained using ANI information or information provided by the subscriber. The updating routine 84 prompts the subscriber to indicate, e.g., by stating "1" to create a new voice dialing entry or "2" to update an existing voice dialing entry what the subscriber wishes to do. The subscriber may be presented with other options as well.

25           If the VD IP 18 detects a "1" response either  
by the use of speech recognition or DTMF detection, the  
subscriber is prompted for the name of the party to be  
added to the voice dialing directory. At this point the  
subscriber states the name and the audio obtained from



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However, if in step 1712 it is determined that a text name entry corresponding to the generated model does not already exist, operation proceeds to step 1714.

However, if in step 1712 it is determined that a text name entry corresponding to the generated model does not already exist, operation proceeds to step 1714.

In step 1714, a text entry for the recognized name is generated, e.g., from information in the dictionary 99.

Then in step 1716, the generated text name, e.g., text entry, is stored in the subscriber's voice dialing record in a manner that associates the text name with the generated SI speech recognition model. From step 1716 operation proceeds to step 1718 wherein the updating sub-routine 1700 is stopped.

10                   In the above described manner, a new voice  
dialing entry can be created from a spoken name with the  
text name and the speech recognition model included in  
the entry being generated from the same audio sample,  
15 e.g., through the use of speech recognition and speech  
recognition model training techniques, respectively. In  
addition, speech recognition models of names which were  
originally generated from text, can be replaced with  
speech recognition models generated from spoken  
20 utterances.

Accordingly, subscriber's can take advantage of the ease of creating voice dialing entries via the Internet and the use of speech recognition models generated from text, while still being able to train speech recognition models via audio samples in cases where the recognition models generated from text provide unsatisfactory recognition results.

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The process of placing a call illustrated in Fig. 18, begins in step 201, with the first subscriber picking up the telephone 38 and dialing \*0, indicating a request for a voice dialing operation. As part of the

In step 202, the CDP feature code trigger set at SSP 4 on the first subscriber's line is hit and call processing is paused. Then, in response to activation of the trigger, the SSP 4 launches, in step 203, a query for call processing instructions to the ISCP 16. The query includes a user ID corresponding to the first subscriber.

In step 204, the ISCP 16 uses service key analysis on the USER ID, included in the received query, and opens the CPR corresponding to the indicated user, i.e., the first subscriber. In accordance with the present invention, the USER ID may be, e.g., the subscriber's Centrex telephone number. Since the feature code trigger \*0 triggered the query, the ISCP 16 knows

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5           In step 212, the SSP 4 routes the call to the  
called party's number in an attempt to connect the caller  
22 with the first called party 26. In step 213, the SSP  
4 detects whether the called party answered or if there  
is a busy, or no answer condition on the called party's  
10 line. If the call is answered, the call is allowed to  
terminate in a normal manner and the ISCP has no further  
involvement in the call.

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In response to the message from the SSP 4, in step 215, the ISCP prepares a second STOR message to be sent to the SSP 4. This time the STOR message includes a Play APP 3 instruction if a busy signal was detected and a Play APP 4 instruction if a no answer condition was detected. The Play App 3 instruction causes the VD IP 18 to play a message indicating that the called number is busy and asking the caller if they want to place another call. It also causes the VD IP 18 to monitor for a



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the call is terminated in step 221 in the normal manner,

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Once connected to the voice dialing IP 18, the voice dialing operation proceeds as previously described in regard to Fig. 18, starting with step 206.

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Provision of voice dialing services to subscribers who are located at remote locations, without access to their remote phone or a phone which has been designated as an easy access phone, are also contemplated. In one such embodiment, access to an individual subscriber is provided by having the subscriber call a telephone number designated for the purpose of providing voice dialing service to remote Centrex subscribers. Different telephone numbers may be designated for providing voice dialing services to

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The steps and signal flows associated with a remote voice dialing service subscriber obtaining voice dialing service are shown in Fig. 20.

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In step 1711, the ISCP 16, accesses the CPR corresponding to the Centrex phone number provided, compares the PIN in the CPR to received PIN. If the stored and received PINs match, the ISCP identifies the address of the VD IP 18 assigned to service the customer and formulates a STOR message with a PLAY APP 1 instruction and the address of the VD IP 18. However, if the ISCP 18 determines the PIN's do not match after three attempts, the ISCP 18 instructs the SSP 2 to terminate the call, e.g., after using the IP 10 to provide the caller with an error message. The transfer of the STOR message or instruction to terminate the call connection generated in step 1711 is transmitted to the SSP 2 in step 1712.

Assuming the caller was validated by the ISCP 16, in step 1713 the caller is coupled, by the SSP 2 to the voice dialing IP 18 indicated by the address included

In step 207, the VD IP 18 recognizes, e.g., based on the USER ID, the subscriber just connected to the VD IP 18 and retrieves the subscriber's voice dialing information, e.g., personal dialing directory information. The retrieved information includes speaker independent speech recognition models of names and telephone numbers corresponding to the modeled names. Once the IP 18 is ready to perform a voice dialing operation, as part of step 207, the VD IP 18 retrieves PROMPT 1 from its memory and plays PROMPT 1 to the subscriber. Prompt 1 is a message which tells the

In step 207, the VD IP 18 recognizes, e.g., based on the USER ID, the subscriber just connected to the VD IP 18 and retrieves the subscriber's voice dialing information, e.g., personal dialing directory information. The retrieved information includes speaker independent speech recognition models of names and telephone numbers corresponding to the modeled names. Once the IP 18 is ready to perform a voice dialing operation, as part of step 207, the VD IP 18 retrieves PROMPT 1 from its memory and plays PROMPT 1 to the subscriber. Prompt 1 is a message which tells the

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subscriber: "State the name of the person you wish to
dial".
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From step 207, the call is ultimately completed  
5 as discussed previously in regard to the Fig. 18 example,  
e.g., by following steps 209 through 221 as may be  
dictated by received input and/or called party  
availability.

10           In the above described manner, a Centrex  
subscriber can initiate a voice dialing operation from a  
specific phone line, e.g., a phone line provisioned with  
Centrex service, a mobile phone, a phone corresponding to  
an "easy access number", or a remote phone with which  
15   Centrex or voice dialing service information has not been  
associated in records maintained by the ISCP 16.

It should be noted that the Fig. 20 call flow assumes that separate IP's are used for security and voice dialing functions. These functions could, if desired, be implemented using the same IP.

Speech recognition models for names have been described as part of a subscriber's voice dialing record. The generation and storage of these speech recognition models, prior to having to perform a speech recognition operation, reduces the amount of processing which must be performed at speech recognition time. However, since the text information from which name models are generated are

stored in the customer record, it is possible to avoid the storage of these models and, instead, generate them when needed from the text information included in a subscriber's voice dialing record.

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Numerous variations on the above described methods and apparatus are possible without departing from the scope of the invention.

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[illegible]